CLAIMS

1. A method for generating a masking threshold level for reducing code quantization in a digital audio system, the threshold comprising both simultaneous masking and temporal masking effects on an audio signal to be coded; the method comprising the steps of:

a) providing a filter having a selected transfer function;

b) inputting simultaneous masking signals into the filter;

c) generating approximate replica temporal masking signals at the filter output;

d) adding the simultaneous masking signals and the replica temporal masking signals to form a composite masking signal; and

e) using the composite masking signal to establish the masking threshold level.

2. The method recited in claim 1 further comprising the steps of:

f) carrying out said code quantization in each of a plurality of frequency domain subbands over a broad audio bandwidth; and

g) performing steps a) through e) in each said subband.

3. The method recited in claim 1 further comprising the steps of:

f) continuously carrying out said code quantization over a plurality of sequential time frames; and

g) performing steps a) through e) over a selected number of said sequential time frames.

4. The method recited in claim 1 wherein said selected transfer function causes said temporal masking signals to decay approximately exponentially with the logarithm of time.

5. The method recited in claim 1 wherein said selected transfer function causes said temporal masking signals to decay at a rate which is approximately inversely proportional to the duration of the corresponding simultaneous masking signal.

6. The method recited in claim 1 wherein said filter is an infinite impulse response filter.

7. The method recited in claim 6 wherein said filter is an M order auto regressive and L order moving average filter.

8. The method recited in claim 7 wherein said filter is selected to have M=2 and L=2.

9. The method recited in claim 1 wherein said selected transfer function is of the form

$$H(z) \approx \frac{Az^{-1} + Bz^{-2}}{1 - Cz^{-1} - Dz^{-2}}$$

where $A \approx .25$, $B \approx 0.06$. $C \approx 0.39$ and $D \approx 0.295$.

10. The method recited in claim 2 wherein step g) is carried out in fewer than the total number of subbands in said plurality of subbands.

11. A method for reducing quantization coding bits in a digital audio system by employing a masking threshold level that includes the effects of both simultaneous masking and temporal masking over a plurality of time frames; the method comprising the steps of:

a) providing a filter which has a selected transfer function for simulating temporal masking decay that is exponential with the logarithm of time;

b) inputting simultaneous masking signals into the filter;

c) generating approximate replica temporal masking signals at the filter output;

d) adding the simultaneous masking signals and the replica temporal

masking signals to form a composite masking signal; and

e) using the composite masking signal to establish the masking threshold level.

12. The method recited in claim 11 further comprising the steps of:

f) carrying out said code quantization in each of a plurality of frequency domain subbands over a broad audio bandwidth; and

g) performing steps a) through e) in each said subband.

13. The method recited in claim 11 further comprising the steps of:

f) continuously carrying out said code quantization over a plurality of sequential time frames; and

g) performing steps a) through e) over a selected number of said sequential time frames.

14. The method recited in claim 11 wherein said selected transfer function causes said temporal masking signals to decay at a rate which is approximately inversely proportional to the duration of the corresponding simultaneous masking signal.

15. The method recited in claim 11 wherein said filter is an infinite impulse response filter.

16. The method recited in claim 15 wherein said filter is an M order auto regressive and L order moving average filter.

17. The method recited in claim 16 wherein said filter is selected to have M=2 and L=2.

18. The method recited in claim 11 wherein said selected transfer function is of the form

$$H(z) \approx \frac{Az^{-1} + Bz^{-2}}{1 - Cz^{-1} - Dz^{-2}}$$

where $A \approx .25$, $B \approx 0.06$. $C \approx 0.39$ and $D \approx 0.295$.

19. The method recited in claim 12 wherein step g) is carried out in fewer than the total number of subbands in said plurality of subbands.

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